



Avaya Solution & Interoperability Test Lab

Application Notes for Configuring SIP Trunking between Qwest iQ® SIP Trunk Service and Avaya IP Office – Issue 1.0

Abstract

These Application Notes describe the steps required to configure Session Initiation Protocol (SIP) trunking between a Qwest iQ® SIP Trunk Service and an Avaya IP Office telephony solution.

Information in these Application Notes has been obtained through DevConnect compliance testing and additional technical discussions. Testing was conducted via the DevConnect Program at the Avaya Solution and Interoperability Test Lab.

1. Introduction

These Application Notes describe the steps to configure Session Initiation Protocol (SIP) trunking between Qwest and an Avaya IP Office telephony solution. The Avaya solution consists of Avaya IP Office, and Avaya H.323 and Digital endpoints. These Application Notes correspond to the Qwest iQ® SIP Trunk Service offered using a Network Border Switch (NBS) in the network.

SIP is a standards-based communications approach designed to provide a common framework to support multimedia communication. RFC 3261 is the primary specification governing this protocol. SIP manages the establishment and termination of connections and the transfer of related information such as the desired codec, calling party identity, etc. Within these Application Notes, SIP is used as the signaling protocol between Avaya IP Office and the Qwest iQ SIP Trunk Service solution.

Customers using this Avaya IP Office telephony solution with the Qwest iQ SIP Trunk Service are able to place and receive PSTN calls via a dedicated broadband Internet connection and the SIP protocol. This converged network solution is an alternative to traditional PSTN trunks such as ISDN-PRI. The text and coverage diagram (**Figure 1**) summarizes the Qwest iQ SIP Trunk Service at the time of writing these Application Notes. Please consult Qwest for the most current description of capabilities.

1.1. Qwest iQ® SIP Trunk

Qwest iQ SIP Trunk enables the origination and termination of local voice, dedicated long-distance, as well as domestic and international toll-free service across a single broadband connection. It is designed to work in conjunction with the Qwest iQ Networking service, which includes a secure, managed, and scalable suite of wide area network (WAN) services.

2. General Test Approach and Test Results

This section describes the interoperability compliance testing used to verify SIP trunk interoperability between the Qwest iQ SIP Trunk Service and an Avaya IP office Telephony Solution.

A simulated enterprise site using an Avaya IP telephony solution was connected to the public Internet using a dedicated broadband connection. The enterprise site was configured to use the commercially available SIP Trunk Service provided by Qwest.

The compliance test included the following:

- Incoming calls to the enterprise site from the PSTN were routed to the DID numbers assigned by Qwest. Incoming PSTN calls were made to H.323 and digital telephones at the enterprise.

- Outgoing calls from the enterprise site were completed via Qwest to PSTN destinations. Outgoing calls from the enterprise to the PSTN were made from H.323, digital, and analog telephones.
- Various call types were tested including: local, long distance, international, inbound/outbound toll-free, emergency, T.38 fax, operator, and directory assistance.
- Calls using G.729A and G.711MU codecs.
- DTMF transmission using RFC 2833 with voice mail for inbound and outbound calls.
- User features such as hold and resume, transfer, and conference.
- Off-net call forwarding and extension to cellular, when the call arrived across the SIP Trunk from Qwest, or when the call forwarding destination and extension to cellular mobile number routed out the SIP Trunk to Qwest, or both.
- Caller ID Presentation and Caller ID Restriction.
- Mobility Features: Mobile twinning to a mobile phone was tested and verified.

2.1. Interoperability Compliance Testing

A simulated enterprise site consisting of an Avaya IP Office telephony solution supporting SIP trunking was connected to the public Internet using a dedicated broadband connection. The enterprise site was configured to use the generally available SIP trunking solution provided by Qwest. This allowed the enterprise site to use SIP trunking for calls to and from the PSTN.

2.2. Test Results

The test objectives were verified. Interoperability testing of the sample configuration was completed with successful results for the Qwest iQ SIP Trunk Service.

2.3. Support

For technical support on the Qwest iQ SIP Trunk Service, contact Qwest Customer Service at <http://www.qwest.com/business/products/bundled-solutions/qwest-iq-sip-trunk/qwest-iq-sip-trunk.html>. Select the Contact a Rep link to send an e-mail inquiry or under the Contact a Rep link select Get Phone Numbers for a list of the support phone numbers available.

3. Reference Configuration

Figure 1 illustrates a sample Avaya IP Office telephony solution connected to Qwest iQ SIP Trunk that was utilized for compliance testing. Since public IP addresses were used during compliance testing, those IP address are not shown in the figure below and are masked (at least partially) throughout the document.

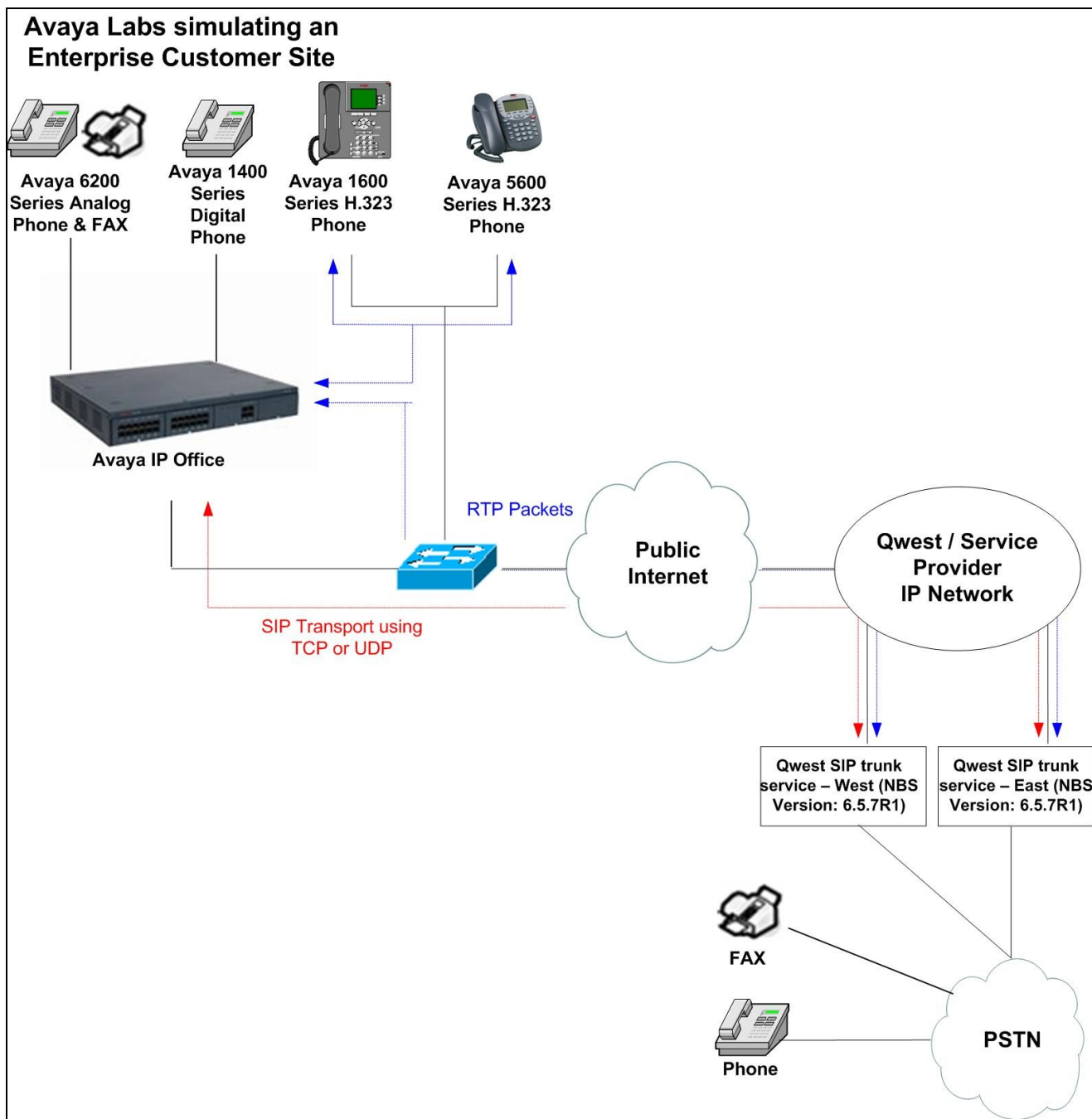


Figure 1: Avaya IP Office Telephony Network connected to Qwest iQ® SIP Trunk

4. Equipment and Software Validated

The following equipment and software were used for the sample configuration provided:

Avaya IP Telephony Solution Components	
Equipment	Software/Firmware
Avaya IP Office 500	6.1 (5)
Avaya IP Office Manager (Windows PC)	8.1 (5)
Avaya IP Office Voicemail Pro	6.1 (15)
Avaya 5610SW IP Telephone (H.323)	2.9.1
Avaya 1616-I IP Telephone (H.323)	1.2.2
Avaya 1416 Digital Telephone	-
Avaya 6219 Analog Telephone and Fax	-
Qwest Solution Components	
Equipment	Software/Firmware
Qwest iQ® SIP Service Network Border Switch (NBS)	6.5.7R1

5. Configure Avaya IP Office

This section describes the steps required for configuring a static SIP trunk on IP Office.

The procedures include the following areas:

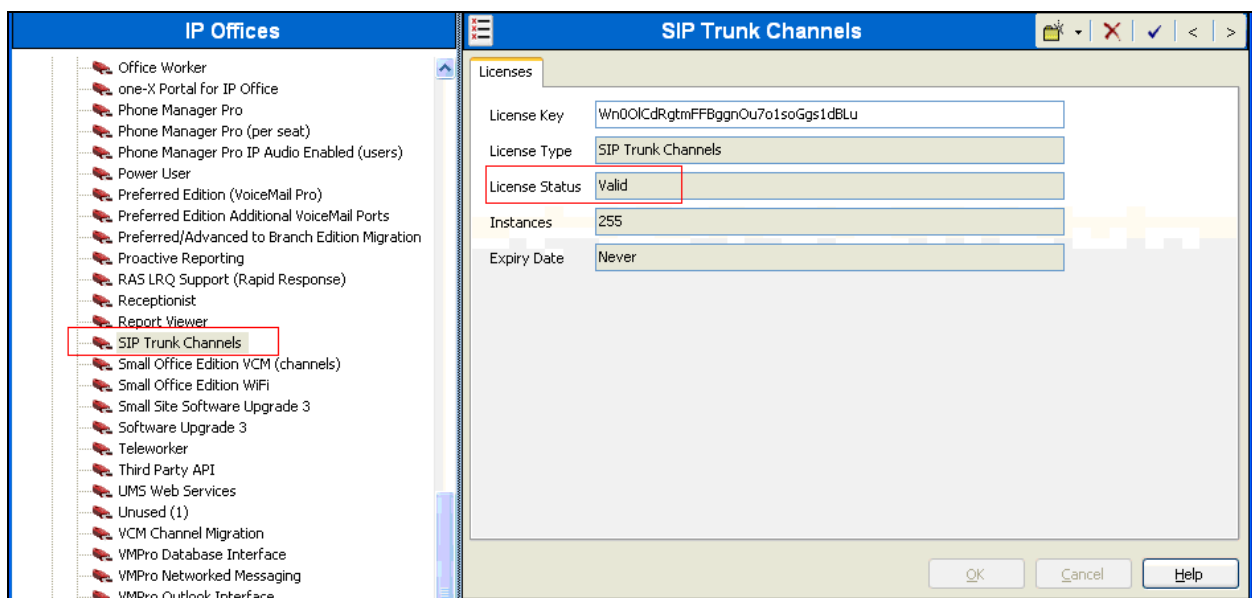
- Verify SIP Trunk Channels License
- Configure WAN interface
- Enable SIP Trunk
- Create the static SIP line for Qwest
- Configure SIP URI parameters for the SIP Line
- Configure T38 Fax parameters for the SIP Line
- Configure SIP OPTIONS timer for “keep alive” function with Qwest
- Configure a short code to route calls to Qwest
- Create an Incoming Call Route for the Inbound SIP calls
- Configure Users’ SIP names

5.1. Verify SIP Trunk Channels License

IP Office is configured via the IP Office Manager application. Log into the PC running the Avaya IP Office Manager application, and select **Start → All Programs → IP Office → Manager** to launch the Manager application. Select the proper IP Office system if there are more than one IP Office system, and log in with the appropriate credentials.

From the configuration tree in the left pane, select **License → SIP Trunk Channels** to display the SIP Trunk Channels screen in the right pane. Verify that the **License Status** field is set to **Valid**.

If a required feature is not enabled or there is insufficient capacity, contact an authorized Avaya sales representative to make the appropriate changes.



5.2. Configure WAN interface

From the configuration tree in the left pane, select **System** to display the System screen in the right pane. Click the **LAN2** tab. Under the **LAN2** tab, select the **LAN Settings** sub-tab, and provide **IP Address** and **IP Mask**.

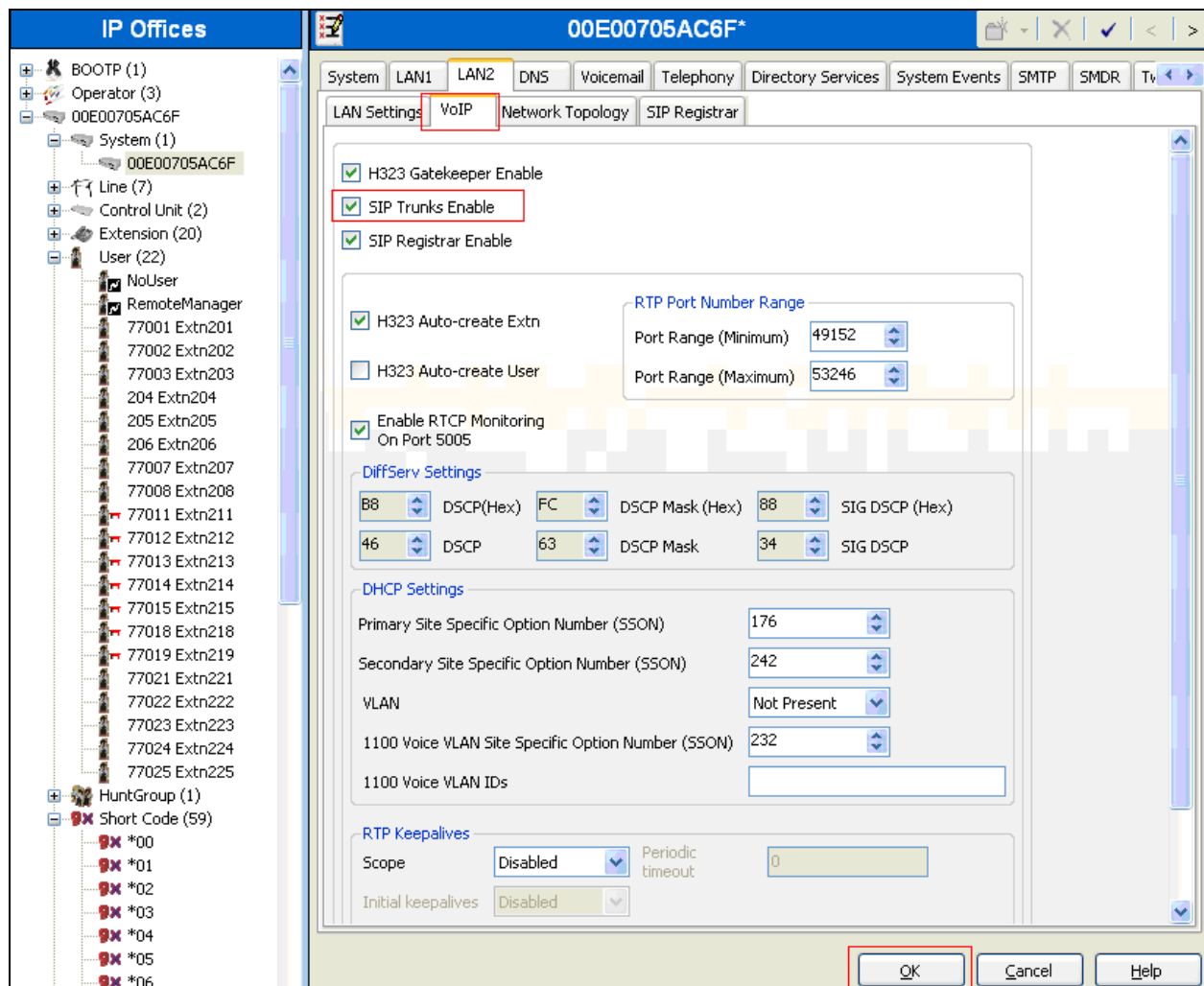
The screenshot displays the Avaya configuration interface for system 00E00705AC6F. The left pane shows the configuration tree with 'System (1)' selected. The right pane shows the 'LAN2' tab with the 'LAN Settings' sub-tab active. The IP Address is set to 205.255.255.85 and the IP Mask is set to 255.255.255.128. Other settings include Primary Trans. IP Address (0.0.0.0), Firewall Profile (<None>), RIP Mode (None), and Number Of DHCP IP Addresses (200). The DHCP Mode is set to Disabled.

Field	Value
IP Address	205 . 255 . 255 . 85
IP Mask	255 . 255 . 255 . 128
Primary Trans. IP Address	0 . 0 . 0 . 0
Firewall Profile	<None>
RIP Mode	None
Enable NAT	<input type="checkbox"/>
Number Of DHCP IP Addresses	200
DHCP Mode	Server <input type="radio"/> Client <input type="radio"/> Dialin <input type="radio"/> Disabled <input checked="" type="radio"/>

5.3. Enable SIP Trunk

From the configuration tree in the left pane, select **System** to display the System screen in the right pane. Click the **LAN2** tab. Under the **LAN2** tab, select the **VoIP** sub-tab, and check the **SIP Trunks Enable** box. Click the **OK** button.

Note: During the initial configuration of Avaya IP Office, the LAN1 was configured as a private network (LAN) and the LAN2 was configured as a public network (WAN). Avaya IP Office can support SIP extensions on the LAN1 and/or LAN2 interfaces, and the compliance test used the LAN2 interface for a SIP trunk termination.



5.4. Create SIP Lines for Qwest

Select **Line** in the left pane. Right-click and select **New → SIP Line**, and create a new **Line Number**. During the compliance test, two SIP lines (21 and 22) were configured. Click the **OK** button.

The screenshot shows the 'SIP Line - Line 22' configuration window with the 'SIP Line' tab selected. The 'Line Number' is set to 22. The 'ITSP Domain Name' is empty. The 'In Service' checkbox is checked. The 'Use Tel URI' checkbox is unchecked. The 'Check OOS' checkbox is checked. The 'Call Routing Method' is set to 'Request URI'. The 'National Prefix' is 0. The 'Country Code' is empty. The 'International Prefix' is 00. The 'Send Caller ID' is set to 'None'. The 'REFER Support' checkbox is unchecked. The 'Incoming' and 'Outgoing' settings are both set to 'Auto'. The 'OK' button is highlighted with a red box.

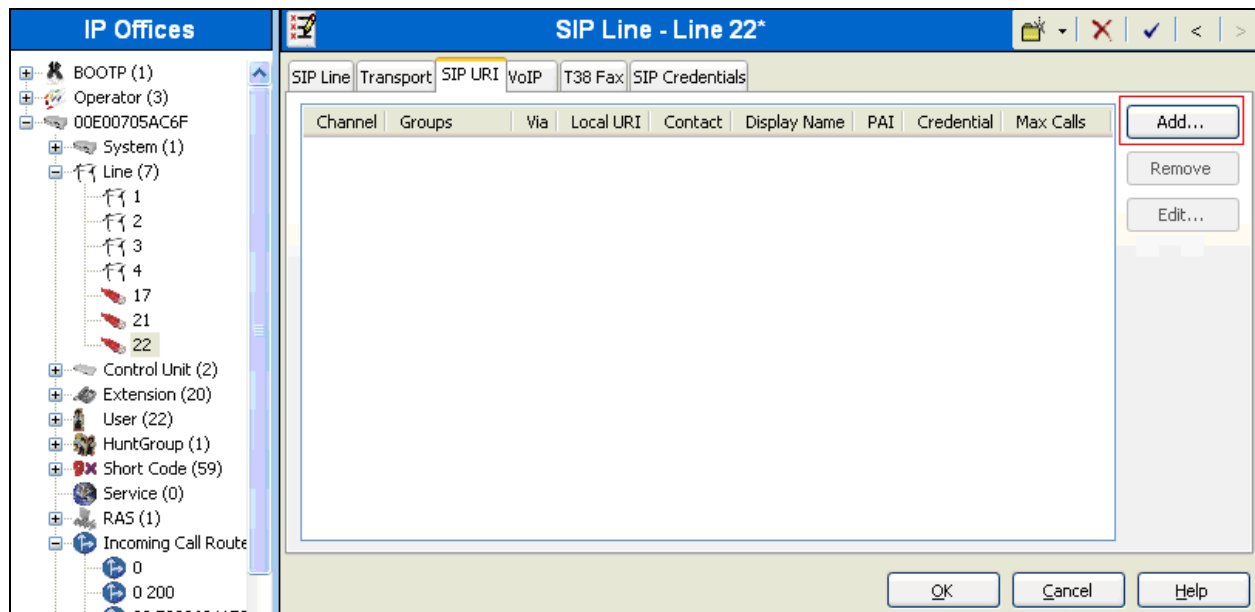
Select the **Transport** sub-tab, and provide the following information:

- **ITSP Proxy Address** – Enter the IP address of the Qwest SIP termination point.
- **Layer 4 Protocol** – Select **UDP**.
- **Use Network Topology Info** – Select **LAN2** (WAN interface).
- Click the **OK** button.

The screenshot shows the 'SIP Line - Line 22' configuration window with the 'Transport' sub-tab selected. The 'ITSP Proxy Address' is 67.171.3.65. The 'Layer 4 Protocol' is set to 'UDP'. The 'Send Port' is 5060. The 'Listen Port' is 5060. The 'Use Network Topology Info' is set to 'LAN 2'. The 'Explicit DNS Server(s)' are 205.171.3.65 and 205.171.2.65. The 'Calls Route via Registrar' checkbox is checked. The 'Separate Registrar' field is empty. The 'OK' button is highlighted with a red box.

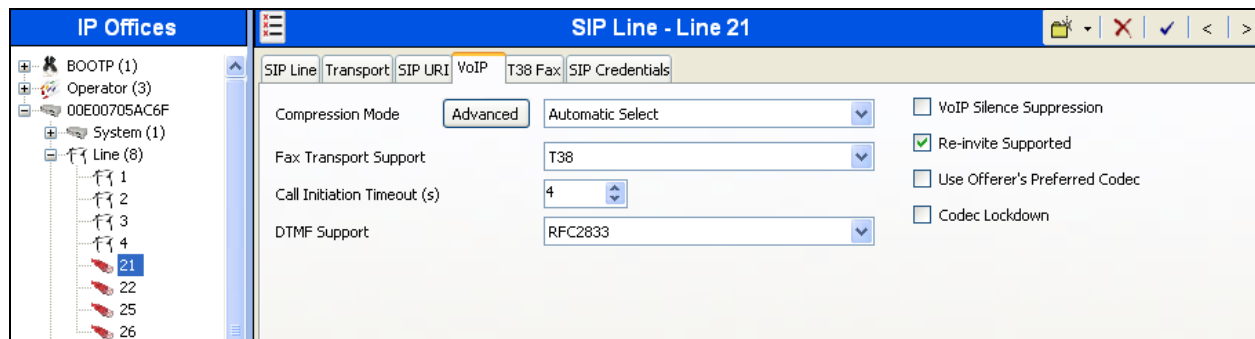
5.5. Configure SIP URI Parameters for the SIP Line

Select the **SIP URI** tab to configure SIP URI parameters for the SIP Line. Click on the **Add** button.



Select **VoIP** to configure the fax configuration. On the **Fax Transport Support** field, select **T38** using the drop down list.

The default values for the other fields may be used



Select **Use Internal Data** for the **Local URI**, **Contact**, and **Display Name** fields. Enter a unique number for the **Incoming Group** and **Outgoing Group** fields. The **Incoming Group** field will be used for mapping inbound calls from the SIP trunk to local stations. The **Outgoing Group** will be used for routing calls externally via the Short Code configured in **Section 5.9**. Use default values for all other fields. Click the **OK** button.

The screenshot shows the 'SIP Line - Line 22*' configuration window. On the left is a tree view of 'IP Offices' containing various system and user entities. The main panel has tabs for 'SIP Line', 'Transport', 'SIP URI', 'VoIP', 'T38 Fax', and 'SIP Credentials'. The 'SIP URI' tab is active, showing an 'Edit Channel' form. The 'Via' field contains '205.185'. The 'Local URI', 'Contact', and 'Display Name' fields are all set to 'Use Internal Data'. The 'PAI' field is set to 'None', and 'Registration' is set to '0: <None>'. The 'Incoming Group' and 'Outgoing Group' fields both contain the value '22'. The 'Max Calls per Channel' is set to '10'. There are 'OK' and 'Cancel' buttons at the top right, and 'OK', 'Cancel', and 'Help' buttons at the bottom right. Red boxes highlight the 'Local URI', 'Contact', 'Display Name', 'Incoming Group', 'Outgoing Group', and the top 'OK' button.

Field	Value
Via	205.185
Local URI	Use Internal Data
Contact	Use Internal Data
Display Name	Use Internal Data
PAI	None
Registration	0: <None>
Incoming Group	22
Outgoing Group	22
Max Calls per Channel	10

5.6. Configure VoIP Parameters for the SIP Line

Select the **VoIP** tab to Configure VoIP parameters for the SIP Line. Click the **Add** button (not shown). For **Compression Mode**, select **Automatic Select** or the desired codec from the drop-down list. Check the **Fax Transport Support** and **Re-invite Supported** check boxes.

Click the **OK** button.

The screenshot shows the 'SIP Line - Line 22' configuration window with the 'VoIP' tab selected. The left sidebar shows a tree view of system components, including 'Line (7)' with sub-items 1 through 22. The main configuration area contains the following settings:

- Compression Mode:** Set to 'Automatic Select' (highlighted with a red box).
- Call Initiation Timeout (s):** Set to '4'.
- DTMF Support:** Set to 'RFC2833'.
- Checkboxes:**
 - ☐ VoIP Silence Suppression
 - ☒ Fax Transport Support
 - ☒ Re-invite Supported
 - ☐ Use Offerer's Preferred Codec

At the bottom right, the 'OK' button is highlighted with a red box, along with 'Cancel' and 'Help' buttons.

5.7. Configure T38 Fax Parameters for the SIP Line

Select the **T38 Fax** tab to configure T38 Fax parameters for the SIP Line, and check the **Use Default Values** check box. Click the **OK** button.

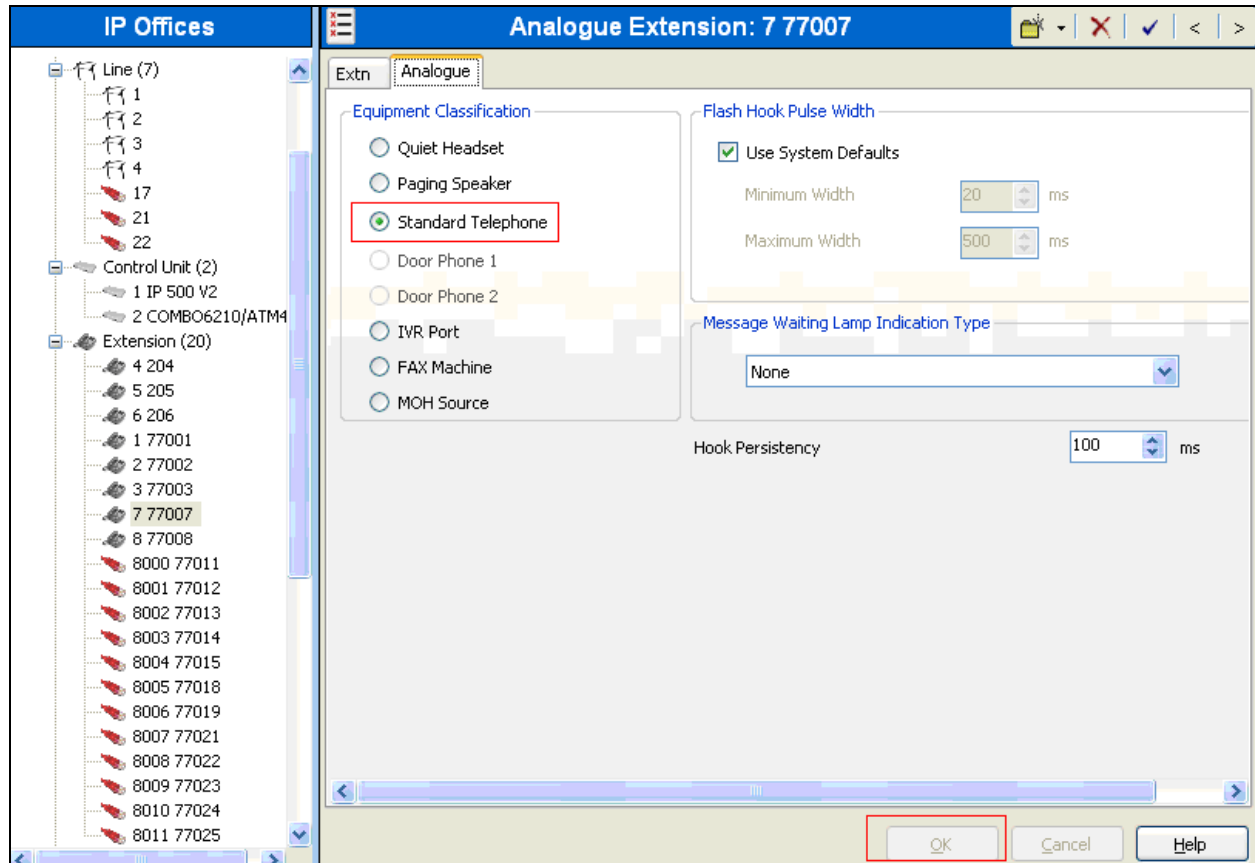
The screenshot shows the 'SIP Line - Line 22*' configuration window with the 'T38 Fax' tab selected. The left sidebar shows a tree view of IP Offices, including BOOTP (1), Operator (3), and various lines. The main area contains the following settings:

- T38 Fax Version: 3
- Transport: UDPTL
- Redundancy: Low Speed (0), High Speed (0)
- TCF Method: Trans TCF
- Max Bit Rate (bps): 14400
- Eflag Start Timer (msecs): 2600
- Eflag Stop Timer (msecs): 2300
- Tx Network Timeout (secs): 150
- ☒ Use Default Values
- Scan Line Fix-up: ☒
- TFOF Enhancement: ☒
- Disable T30 ECM: ☐
- Disable EFlags For First DIS: ☐
- Disable T30 MR Compression: ☐
- NSF Override: ☐
- Country Code: 0
- Vendor Code: 0

The 'OK' button is highlighted with a red box.

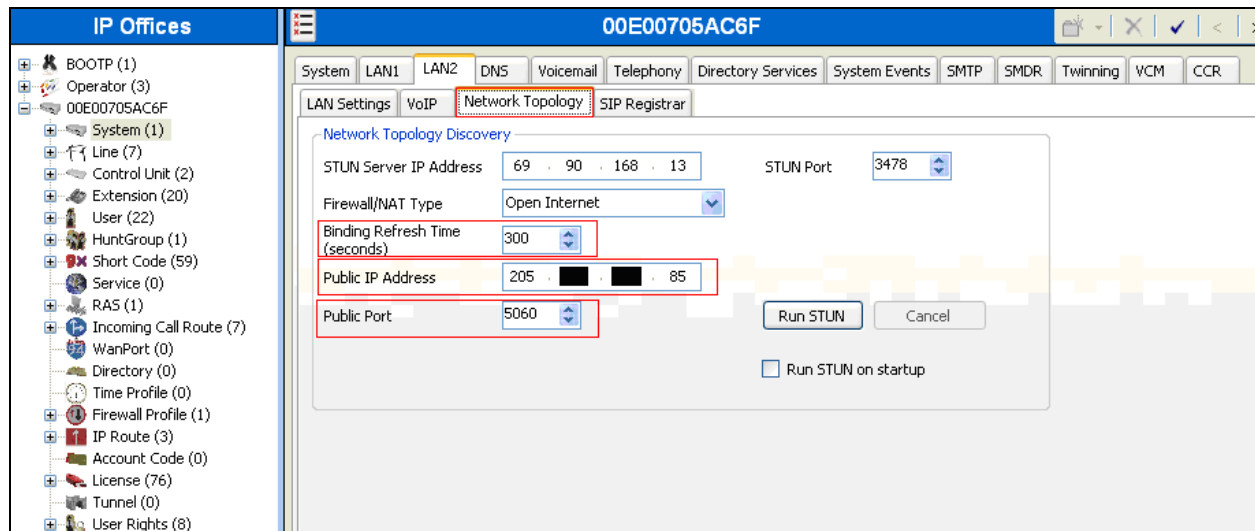
A FAX machine was connected to an analog extension. The following screen shows the analog extension that was configured as a FAX line.

Set the Equipment Classification field as **Standard Telephone**. Click the **OK** button.



5.8. Configure SIP OPTIONS Timer for “keep alive” Function with Qwest

Select **System** in the left pane. Under the **LAN2** tab, select the **Network Topology** sub-tab. Set the **Binding Refresh Time** to the desired interval which determines the frequency with which OPTIONS messages will be sent to Qwest. For this compliance testing, it is set to 300. The default value of 0 means disabling OPTIONS messages. For **Public IP Address**, enter the Avaya IP Office system public IP address (WAN). Confirm that **Public Port** is set to 5060 and accept the default values for all other fields. Click the **OK** button (not shown).



5.9. Configure a Short Code to Route Calls to Qwest

Select **Short Code** in the left panel. Right click and select **Add**. Enter [x]N;, where [x] is a valid number, in the **Code** text box. The number 9 is used for [x] in the example below. This code requires the user to dial the digit 9 followed by the destination's telephone number symbolized by N in order to route the call out the SIP Trunk.

Note: N can be any number other than a local IP Office extension. For example, a 10-digit number, a toll-free number, directory assistance, information service, etc.

Select **Dial** for the **Feature**. Enter the **Outgoing Group** number created in **Section 5.5** for the **Line Group Id** field. Enter the dialed number N followed by “@<IP address of Qwest>” for the **Telephone Number** field. The **Telephone Number** field is used to construct the SIP URI in the **To** field of the outgoing SIP INVITE message. Use default values for all other fields. Click the **OK** button.

The screenshot displays the 'IP Offices' configuration window. On the left, a list of short codes is shown, including *37*N#, *38*N#, *39, *40, *41, *42, *43, *44, *45*N#, *46, *47, *48, *49, *50, *51, *52, *53*N#, *57*N#, *70*N#, *71*N#, *9000*, *91N;, *92N;, *DSSN, *SDN, *SKN, 0N, 2200x, and 9N;. The '9N;' code is selected. The main configuration area on the right is titled '9N;; Dial'. It contains the following fields: 'Code' (9N;), 'Feature' (Dial), 'Telephone Number' (N"@67. [redacted].8"), 'Line Group Id' (22), 'Locale' (United States (US English)), and 'Force Account Code' (unchecked). The 'OK' button is highlighted with a red box.

5.10. Create an Incoming Call Route for the Inbound SIP Calls

Select **Incoming Call Route** in the left pane. Right-click and select **New**.

Enter the following:

- **Any Voice** for the **Bearer Capability** field.
- The **Incoming Group** number created for the URI in **Section 5.5** in the **Line Group Id** field.
- The 10-digit DID number provided by Qwest that is mapped back to a local IP Office extension in the **Incoming Number** field.
- Use default values for all other fields.

The screenshot shows the 'Incoming Call Route' configuration window for the number 22 7203624173. The 'Standard' tab is selected. The following fields are visible:

Field	Value
Bearer Capability	Any Voice
Line Group Id	22
Incoming Number	7203624173
Incoming Sub Address	
Incoming CLI	
Locale	
Priority	1 - Low
Tag	
Hold Music Source	System Source

The 'OK' button is highlighted with a red box.

- Next, navigate to the **Destinations** tab and select the desired local extension number from the drop down list.
- Click the **OK** button.

The screenshot shows the 'Incoming Call Route' configuration window for the number 22 7203624173, with the 'Destinations' tab selected. The following table is visible:

TimeProfile	Destination	Fallback Extension
Default Value	77021 Extn221	

The 'OK' button is highlighted with a red box.

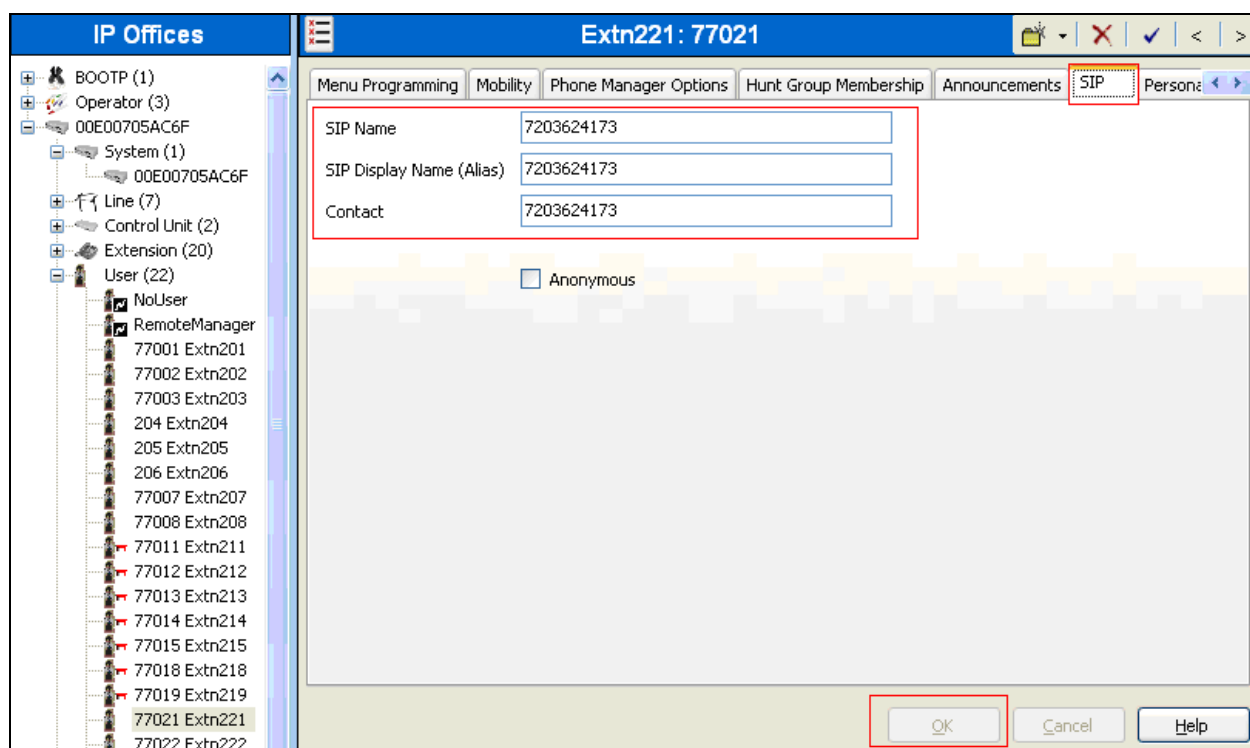
5.11. Configure Users' SIP Names

Select **User** in the left panel. Select the desired user by double-clicking on an entry in the left panel. Select the **SIP** tab.

Modify the **SIP Name** and **Contact** fields to the DID number provided by Qwest that is used for this particular extension. These settings instruct the system to use this DID number to construct the:

- user part of the SIP URI in the From header of an outgoing SIP INVITE message
- user part of the SIP URI in the Contact header of an outgoing SIP INVITE message

Modify the **SIP Display Name (Alias)** that will be used for the SIP Display info. The other fields can be left as defaults. Click the **OK** button.



After making the changes, click on the floppy disk icon (not shown) to push the changes to the IP Office system and have them take effect

Note: *Changes will not take effect until this step is completed. This may cause a reboot of Avaya IP Office causing service disruption.*

6. Configure Qwest

To use the Qwest iQ SIP Trunk Service, a customer must request service from Qwest using their sales processes. The process can be started by contacting Qwest via the corporate web site at <http://www.qwest.com> and requesting information via the online sales links or telephone numbers.

7. Verification Steps

This section provides verification steps that may be performed to verify that the Avaya H.323, digital and analog endpoints can place outbound calls and receive inbound calls through Qwest iQ SIP Trunk Service solution.

- Verify that endpoints at the enterprise site can place calls to the PSTN and that the call remains active for more than 35 seconds. This time period is included to verify that proper routing of the SIP messaging has satisfied SIP protocol timers.
- Verify that endpoints at the enterprise site can receive calls from the PSTN and that the call can remain active for more than 35 seconds.
- Verify that the user on the PSTN can terminate an active call by hanging up.
- Verify that an endpoint at the enterprise site can terminate an active call by hanging up.

8. Conclusion

These Application Notes describe the configuration steps required to connect customers using an Avaya IP Office telephony solution to a Qwest iQ SIP Trunk Service solution. The Qwest iQ SIP Trunk Service is a SIP-based Voice over IP solution.

9. Additional References

This section references the documentation relevant to these Application Notes. Additional Avaya product documentation is available at <http://support.avaya.com>.

- [1] *IP Office 6.0 Installation, Issue 21j*, April 2010, Document Number 15-601042
- [2] *IP Office Release 6 Manager 8.0, Issue 24k*, April 2010, Document Number 15-601011
- [3] RFC 3261 *SIP: Session Initiation Protocol* <http://www.ietf.org/>
- [4] RFC 2833 *RTP Payload for DTMF Digits, Telephony Tones and Telephony Signals* <http://www.ietf.org/>

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